ECE540: Homework #5

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Chapter 9 (7^{th} Edition)

Review Questions

Section 9.2

1. R6. List three disadvantages of UDP streaming.

   - UDP doesn't transmit at constant rate or with continuous playout.
   - A media control server is required, creating larger overhead and more complexity.
   - Many firewalls do not allow UDP.

Section 9.3

2. R9. What is the difference between end-to-end delay and packet jitter? What are the causes of packet jitter?

   End-to-end delay is the amount of time it takes a packet to go from source to destination. It is the accumulation of transmission, processing, and queuing delays in routers; propagation delays in links; and end-system processing delays. Packet jitter is the fluctuation in end-to-end delays between packets. Congestion, low bandwidth links, and changing paths can all cause packet jitter.

3. R10. Why is a packet that is received after its scheduled playout time considered lost?

   Packets have a limited time window, if a packet arrives after this window (playout time), it will expire and be discarded. Since it is discarded, it will be considered lost.

Section 9.4

4. R12. How are different RTP streams in different sessions identified by a receiver? How are different streams from within the same session identified?

   - RTP streams have multicast addresses and provide one-to-many and many-to-many multicast trees.
   - A Synchronization Source Identifier is sent with each stream that identifies streams within the same session.

Problems

1. P2. Recall the simple model for HTTP streaming shown in Figure 9.3. Recall that B denotes the size of the client's application buffer, and Q denotes the number of bits that must be buffered before the client application begins playout. Also r denotes the video consumption rate. Assume that the server sends bits at a constant rate x whenever the client buffer is not full.

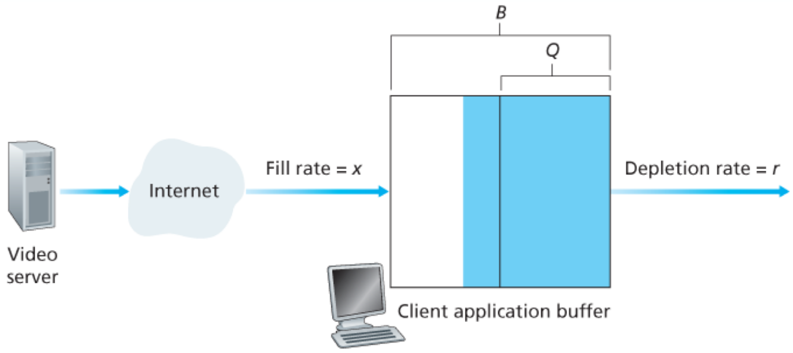


Figure 9.3: Analysis of client-side buffering for video streaming

(a) Suppose that $x < r$. As discussed in the text, in this case playout will alternate between periods of continuous playout and periods of freezing. Determine the length of each continuous playout and freezing period as a function of Q, r, and x.

Since the server cannot send bits as quickly as the video is consuming them, the continuous playout and freezing period will be bottle-necked at $\frac{Q}{x}$

(b) Now suppose that $x > r$. At what time $t = t_f$ does the client application buffer become full?

Since the server is sending bits quicker than the video can consume, the buffer will fill up. This will happen at a rate of $\frac{Q}{x} + \frac{B-Q}{x-r}$

2. P6. In the VoIP example in Section 9.3, let h be the total number of header bytes added to each chunk, including UDP and IP header.

(a) Assuming an IP datagram is emitted every 20 msecs, find the transmission rate in bits per second for the datagrams generated by one side of this application.

$$\text{datagram} = 160 \text{ bytes} + h \qquad \# \text{ datagrams} = \frac{1}{20 \text{ msecs}} = 50$$

$$\begin{aligned} r_{\text{trans}} &= 50 \cdot 8 \text{ bits} \cdot (160 \text{ bytes} + h) \\ &= 64 + 0.4h \text{ kbps} \end{aligned}$$

(b) What is a typical value of h when RTP is used?

The typical value is the UDP header + IP header + RTP header, so 40 bytes.

3. P10. Compare the procedure described in Section 9.3 for estimating average delay with the procedure in Section 3.5 for estimating round-trip time. What do the procedures have in common? How are they different?

   Estimating average delay and estimating round-trip time both use the same formula and both use previously calculated delays. RTT calculates the entire trip time, while average delay only calculates source to destination (not including queueing delays). For RTT, each segment is count, for estimating average delay, only counted once it has been received.

4. P15.

   (a) Suppose we send into the Internet two IP datagrams, each carrying a different UDP segment. The first datagram has source IP address A_1, destination IP address B, source port P_1, and destination port T. The second datagram has source IP address A_2, destination IP address B, source port P_2, and destination port T. Suppose that A_1 is different from A_2 and that P_1 is different from P_2. Assuming that both datagrams reach their final destination, will the two UDP datagrams be received by the same socket? Why or why not?

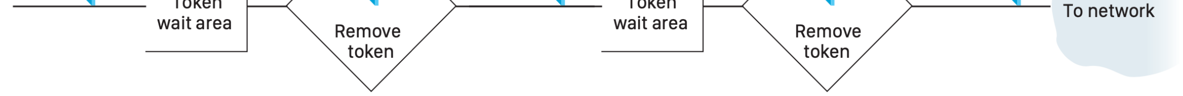
   Yes, both UDP datagrams will be received by the same socket since both have the same destination IP and port. The receiver has a demultiplexer to resolve these.

   (b) Suppose Alice, Bob, and Claire want to have an audio conference call using SIP and RTP. For Alice to send and receive RTP packets to and from Bob and Claire, is only one UDP socket sufficient (in addition to the socket needed for the SIP messages)? If yes, then how does Alice's SIP client distinguish between the RTP packets received from Bob and Claire?

   Yes, one UDP socket is sufficient. Each of Bob and Alice's RTP packets will have their own IP addresses.

5. P20. Consider the leaky bucket policer that polices the average rate and burst size of a packet flow. We now want to police the peak rate, p, as well. Show how the output of this leaky bucket policer can be fed into a second leaky bucket policer so that the two leaky buckets in series police the average rate, peak rate, and burst size. Be sure to give the bucket size and token generation rate for the second policer.

   Please see modified figure below. Bucket size = 1 token, token generation rate = $p \ \frac{\text{tokens}}{\text{sec}}$.


r tokens/sec
Bucket holds up to b tokens
p tokens/sec
Bucket holds 1 token
Packets
Token wait area
Remove token
Token wait area
Remove token
To network